Necessary Technical and Operational Elements of a VoIP Interconnection[1] Agreement

Introduction

Currently, the interconnection agreements (ICAs) that exist between the RBOCs and competitive service providers contain specific provisions for the exchange of traffic using TDM. These agreements are governed by the basic provisions of sections 251/252 that require an incumbent local exchange carrier to interconnect with a competitor and agree to the reciprocal exchange of traffic, even where the competitor may be significantly smaller. This 251/252 framework mandates good-faith negotiations between incumbents and competitive carriers, including protections (such as the public filing of agreements and state commission approval) designed to prevent discrimination and promote competition. These statutory provisions are technology neutral and provide the framework as new interconnection and transport technologies emerge to replace the old. The purpose of this analysis is to identify the key parameters that can and should be addressed in amendments to current interconnection agreements to accommodate VoIP interconnection.[2]

As an initial matter (as noted in Attachment B), amendments to existing agreements to exchange, transport and terminate traffic using new technology should not consider the end-point technology used by subscribers of either party (i.e. TDM or IP); rather, the agreement should facilitate the use of VoIP interconnection and transport for <u>all</u> voice traffic exchanged between the parties. Second, the amendments to existing interconnection agreements should address VoIP interconnection by building the VoIP inter-operability necessary for the actual exchange of traffic upon a well-defined physical interconnection of the parties' <u>managed</u> IP

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[1] VoIP interconnection is a term used within this document to mean the facilities-based interconnection of carriers' managed IP networks for the purpose of exchanging PSTN traffic.

[2] The existing agreements cover necessary terms and conditions for interconnection with an RBOC in general. The purpose of the amendment would be to identify with specificity the unique issues presented by VoIP interconnection.

networks. As explained in previous filings,[3] managed IP networks are capable of providing the deterministic performance necessary to support the service level requirements of the PSTN; something of which the Internet is incapable.

Just as for TDM, there are several parameters that must be specified for VoIP interconnection. These parameters should be set forth in the interconnection agreement's amendment to accommodate the operational and technical aspects of VoIP interconnection.[4] A VoIP interconnection agreement between competitive providers and RBOCs should include, at a minimum, the following elements:[5]

Locations for Points of VoIP Interconnection
Supported Media Types, QoS Parameters, CODEC Transcoding, Bandwidth Requirements, etc.
Specifications for Physical Interconnection (Layer-1 & Layer-2)
Network Reliability and Security Policy for External Network-to-Network Interfaces
Network Support Practices and Infrastructure Inter-operability for Emergency Services
OSS Procedures
Interconnection Establishment/Activation Procedures
Fallback Procedures
KPI Measurement/Management/Oversight/Reporting of Service Level Covenants

These elements (and their importance) are described below.

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[3] See Comments of COMPTEL, *In the Matter of Facilitating the Deployment of Text-to-911 and Other Next Generation 911 Applications, Framework for Next Generation 911 Deployment*, PS Docket Nos. 11-153, 10-255, Attachment, "IP INTERCONNECTION FOR MANAGED VOIP" April, 2011, at 21-22 (filed Dec. 12, 2011).

[4] Compensation issues need to be addressed in the ICA as well, but are outside the scope of this analysis.

[5] The granularity of agreements between *willing* partners is likely to provide less detail than must be spelled out in an interconnection agreement with an RBOC that is opposed to the competitive rights and protections of section 251/252. Consequently, we would expect more detail being included in interconnection agreements with RBOCs to limit the range of potential future disputes.

Exploring the Elements

LOCATIONS FOR POINTS OF INTERCONNECTION

RBOCs deploy Network Border Elements[6] (ex. Session Border Controller) at different places in their IP networks where they expect to exchange traffic with an external party. These may be in the core network or in the subscriber edge network. The role of these elements is to provide the physical interface necessary to achieve interconnection for the provision of VoIP services. For example, shown in Figure 1 is an illustration prepared by AT&T to show how Border Elements and Network Gateway Border Elements are deployed by AT&T to provide its IP Flexible Reach service.[7]

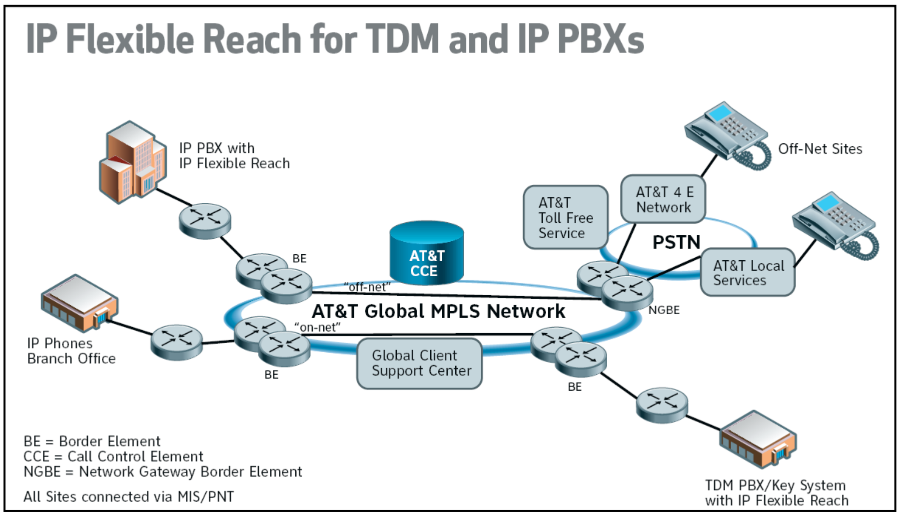

Figure 1. – AT&T service provided via Border Elements to its managed (MPLS) IP network.[8]

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[6] A Network Border Element is a device or function which provides interconnection to external networks. It may be configured to provide isolation and protection from external interconnected networks including those of end users and interconnected service providers.

[7] IP Flexible Reach Service is a retail product offered by AT&T. It is a SIP trunking service that delivers integrated access for IP PBX, TDM PBX or Key System environments to subscribers over AT&T's managed IP network. IP Flexible Reach is, of course, a "service" and not a suitable surrogate for intercarrier interconnection because it is not capable of supporting cross-functional requirements of intercarrier interconnection (i.e. exchange access, multi-jurisdictional call origination/termination, SS7 signaling elements within SIGTRAN, database access, etc.). However, it uses the same Border Elements that, configured for the purpose, *would* be capable of serving as points of interconnection for intercarrier VoIP traffic exchange.

[8] See AT&T IP Flexible Reach brochure available at http://www.business.att.com/content/productbrochures/IP_Flexible_Reach.pdf .

These Border Elements provide interconnection to AT&T's managed IP network[9] for subscribers whose voice service is packet-based, while the Network Gateway Border Elements provide TDM to IP (and IP to TDM) transcoding for network-to-network interconnection to AT&T's TDM network in order to reach AT&T's TDM subscribers and the subscribers of other carriers.

By definition, each of these Border Elements represents a technically feasible point of interconnection to the RBOC's managed IP network for the purpose of exchanging voice traffic, and interconnection with these elements should be available to any requesting carrier for that purpose. Further, the RBOCs' managed IP network itself, to the extent it maintains a point of presence within a carrier hotel (or similar location) for the purpose of exchanging traffic with external parties, presents a technically feasible point of VoIP interconnection and should be available to any requesting carrier for interconnection. Indeed, it is reasonable to expect that many requesting carriers will *prefer* interconnection at neutral carrier hotels that have been developed precisely for the purpose of interconnection.

When selecting the subset of these available points of interconnection for actual use, the requesting carrier will desire (as should the RBOC) an interconnection configuration that efficiently supports the specific volume and geographic nature of the traffic flows between them. This may cause, in the case of a national competitor for example, for the requesting carrier to seek a nationalized configuration, using three or four interconnection points to serve all domestic US traffic. Conversely, it could result in an agreement with a small regional competitor, where the parties exchange traffic at two interconnection points within a single metropolitan area.

The efficiency of VoIP interconnection and transport is maximized when all technically feasible points of interconnection are considered in the design of an interconnection configuration. Traffic can then be concentrated to fewer facilities and interconnection ports which more closely fit the traffic flow's origin and destination. This objective is best achieved when all Network Border Elements are available for interconnection.

The interconnection agreement should provide for traffic forecasts from both the competitor and the RBOC to be used for the purpose of determining the most efficient network topology of initially selected points of interconnection. It should also provide for the consideration of subsequently available points, as introduced, in order to maximize the efficiency of interconnection between the parties.

[9] MPLS is an acronym for Multiprotocol Label Switching, a managed network protocol suite used to support service level guarantees for network performance. See IETF RFC 3031 "Multiprotocol Label Switching Architecture" available at https://tools.ietf.org/html/rfc3031 .

SUPPORTED MEDIA TYPES, QOS PARAMETERS, CODEC TRANSCODING AND BANDWIDTH REQUIREMENTS

Though IP technology can support the transmission of multiple types of real-time media (ex. – voice, video, real-time text, presence, etc.), the media type that must be considered necessary in a VoIP interconnection agreement for current PSTN traffic is voice (audio). Therefore, it will be the expected voice traffic that must be analyzed to create the bandwidth profiles necessary for the sizing and design of interconnection facilities and ports.

Moreover, the appropriate minimal measures of service quality should include those that indicate the quality of the audio transmission and those that reflect the signaling and call handling capacity of the network. Mean Opinion Score (MOS) is the measure of audio quality in the PSTN.[10] Call handling capacity and performance is measured using metrics such as Answer Seize Ratio (ASR), Post Dial Delay (PDD) and Network Efficiency Ratio (NER) among others.[11]

In VoIP interconnection and transport the call quality measure, pMOS,[12] represents the predicted effect of all network performance characteristics on the quality of the audio as received by the end-user. These network performance characteristics include latency, jitter and packet loss and are principally reflective of the network's capacity to transport packets of information in a reliable and consistent fashion.

Because the call quality measurement on the PSTN is MOS, and the accepted measure of high call quality on the PSTN is generally accepted to reflect a MOS score of 4.0 – 4.4[13] (out of a scale maximum of 5), a MOS score of 4.0 should be defined as the minimally acceptable pMOS (i.e. predicted MOS) call quality threshold, as determined from the point of interconnection to each

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[10] See ITU-T Recommendation P.800.1 "Mean Opinion Score (MOS) terminology" (07/2006)

[11] See, for example, ITU-T Recommendation E.411 "International Network Management – Operational Guidance" and E.422 "Quality of Service for Outgoing International Calls" available at http://www.itu.int/en/ITU-T/publications/Pages/recs.aspx

[12] pMOS, a measure of call quality for VoIP service, is an acronym for predicted Mean Opinion score. ITU-T Recommendation P.564 provides the method for deriving pMOS.

[13] See, for example, maximum achievable MOS score with G.711u CODEC in Alcatel-Lucent document "Delivering Voice Services with Alcatel-Lucent's Triple Play Service Delivery Architecture (TPSDA)" available at http://www.google.com/url?sa=t&rct=j&q=mean%20opinion%20score%20pstn%20toll%20alcatel&source=web&cd=1&ved=0CDcQFjAA&url=http%3A%2F%2Fwww.alcatel-lucent.com%2Fwps%2FDocumentStreamerServlet%3FLMSG_CABINET%3DDocs_and_Resource_Ctr%26LMSG_CONTENT_FILE%3DBrochures%2FVoice_overTPSDA_an_v2.pdf&ei=-G3JUYyXNqbT0wG4mIG4Aw&usg=AFQjCNFnw9ivAmE3ivLm2MoOmY175digOA

party's respective end-user, using the common CODEC of the PSTN (G.711).[14] By using pMOS as the measure of call quality, the summary effect of other, more granular parameters such as crosstalk, distortion, echo, echo canceller performance, fading, latency, loudness, jitter, noise, packet loss and silence suppression/voice activity detection (VAD) performance can be subsumed.

CODECs transcode voice to digital information (and back) accomplishing varying levels of efficiency based on signal sampling rates, voice payload size and signal loss. To determine the types and configuration of CODECs to be supported at a VoIP interconnection interface, the agreement should provide for maximum flexibility, constrained only by the limitations of the involved equipment. Various Session Description Protocol (SDP)[15] profiles will be defined at an operational level, which will determine the actual CODECs available for SDP offer/answer negotiation in any specific call, however, support for long-standing as well as newer CODECs should be available in the interconnection agreement.

At a minimum, support for the G.711 CODEC should be mandated for use. In addition, support for the wideband CODEC[16] and the common audio compression CODEC[17] should be provided as well. Any other CODEC could be deployed by mutual agreement.

Figure 2 shows the bandwidth consumed for sustainable voice packets created using the associated CODEC as calculated by Cisco.[18] Additional bandwidth will be required for SIP signaling and for network management protocols, which are specific to the technologies chosen for network management (ex. MPLS, Carrier Ethernet, etc.).

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[14] The common CODEC of the PSTN is the CODEC standardized in ITU-T Recommendation G.711 and is often referred to simply as "G.711".

[15] Session Description Protocol is used for negotiation between end points of media type, format, and other associated properties. See IETF RFC 4566 "SDP: Session Description Protocol " available at https://tools.ietf.org/html/rfc4566

[16] See ITU-T Recommendation G.722.

[17] See ITU-T Recommendation G.729.

[18] See Cisco Document #7934 "Voice Over IP - Per Call Bandwidth Consumption", updated February 2, 2006 available at http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a0080094ae2.shtml

Codec Information				Bandwidth Calculations					
Codec & Bit Rate (Kbps)	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	Mean Opinion Score (MOS)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Packets Per Second (PPS)	Bandwidth MP or FRF.12 (Kbps)	Bandwidth w/cRTP MP or FRF.12 (Kbps)	Bandwidth Ethernet (Kbps)
G.711 (64 Kbps)	80 Bytes	10 ms	4.1	160 Bytes	20 ms	50	82.8 Kbps	67.6 Kbps	87.2 Kbps
G.729 (8 Kbps)	10 Bytes	10 ms	3.92	20 Bytes	20 ms	50	26.8 Kbps	11.6 Kbps	31.2 Kbps
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	3.9	24 Bytes	30 ms	33.3	18.9 Kbps	8.8 Kbps	21.9 Kbps
G.723.1 (5.3 Kbps)	20 Bytes	30 ms	3.8	20 Bytes	30 ms	33.3	17.9 Kbps	7.7 Kbps	20.8 Kbps
G.726 (32 Kbps)	20 Bytes	5 ms	3.85	80 Bytes	20 ms	50	50.8 Kbps	35.6 Kbps	55.2 Kbps
G.726 (24 Kbps)	15 Bytes	5 ms		60 Bytes	20 ms	50	42.8 Kbps	27.6 Kbps	47.2 Kbps
G.728 (16 Kbps)	10 Bytes	5 ms	3.61	60 Bytes	30 ms	33.3	28.5 Kbps	18.4 Kbps	31.5 Kbps
G722_64k(64 Kbps)	80 Bytes	10 ms	4.13	160 Bytes	20 ms	50	82.8 Kbps	67.6Kbps	87.2 Kbps
ilbc_mode_20 (15.2Kbps)	38 Bytes	20 ms	NA	38 Bytes	20 ms	50	34.0Kbps	18.8 Kbps	38.4Kbps
ilbc_mode_30 (13.33Kbps)	50 Bytes	30 ms	NA	50 Bytes	30 ms	33.3	25.867 Kbps	15.73Kbps	28.8 Kbps

Figure 2. - Cisco estimates of bandwidth consumption for common CODECs

The interconnection agreement can specify that parties should allocate, on a percentage basis, the amount of call volume expected for each supported CODEC. Using the parameters mentioned above, along with the estimations of signaling traffic volume required for the call volume as forecast in the first step, a bandwidth profile for each CODEC should be defined. These bandwidth profiles can then be used to determine the bandwidth required for the interconnection facilities described below.

SPECIFICATIONS FOR PHYSICAL INTERCONNECTION (LAYER-1 & LAYER-2)

As discussed previously, the physical point of interconnection for VoIP interconnection between an RBOC and competitive service provider will, most likely, occur between the Network Border Elements,[19] or managed IP network elements, of the respective parties. The Layer-1 (physical) interface ports of these devices can be provided using optical or electrical interface technologies. Although there are options available to support Layer-2 protocols other than Ethernet, Ethernet is ubiquitous and should represent the primary option for the VoIP interconnection data link level (Layer-2) interface.

[19] A device or function which protects and hides the internal network from external entities to which it interconnects (Ex. - Session Border Controller)

In determining the capacity of the physical ports used for VoIP interconnection, the parties will require the projected traffic volumes at each point of interconnection. Using this data, the parties can determine, for example, whether an optical interface is warranted or if an electrical interface will suffice. Electrical interfaces can support Ethernet speeds of up to 1Gb/s, while optical interfaces, though more expensive, can be configured to support far greater Ethernet speeds (often up to 10Gb/s).[20] Both RBOCs and competitive service providers should provide electrical interfaces at Ethernet speeds of 100/1000 Mb/s and, where the requested capacity warrants, optical interfaces at speeds of 1 Gb/s and 10 Gb/s.

Once port and bandwidth parameters are defined, the parties can then define the signaling and call handling through-put requirements, or Committed Capacity,[21] for each interconnected port. Committed Capacity is specified by two parameters: first, a parameter specifying the maximum number of calls (sessions) that may be initiated within a finite time interval (ex. – per second); and second, a parameter specifying the maximum number of concurrent calls that can be supported on each interconnected port.

By using the Committed Capacity parameters for each point of interconnection, along with the expected percentage of that traffic forecast for each bandwidth profile specified in the associated project (i.e. Determining Supported Media Types, QoS Parameters, CODEC Transcoding and Bandwidth Requirements), the parties can determine the bandwidth requirement for the various information flows that will transit each interconnected port. This bandwidth requirement can then be specified in the Layer-2 parameters of each information flow as a Committed Information Rate and included in the interconnection agreement.

NETWORK RELIABILITY AND SECURITY POLICY FOR EXTERNAL NETWORK-TO-NETWORK INTERFACES

The engineering practices that describe the process used to design survivability and load-balancing routes, facilities and equipment in order to maintain high-reliability services such as those of the PSTN are well established. Though the specific nature of these practices depend upon the technology in question, almost always they require the deployment of redundancy as a main-stay of carrier-grade availability. Figure 3 shows the 3-tier hierarchical model "typically employed to achieve a high performance, highly available, scalable network design. This design employs the four key design principles of hierarchy, modularity, resiliency and flexibility."[22]

[20] See, for example, Acme Packet Net-Net 9200 information available at http://www.acmepacket.com/collateral/acm/datasheet/APKT_DS_NetNet9200.pdf

[21] Committed Capacity is defined as the call handling capacity made available by each carrier at any IP-to-IP meet point arrangement.

[22] See Cisco 3-tier Hierarchical Model discussion available at http://www.cisco.com/en/US/docs/solutions/Enterprise/Education/SchoolsSRA_DG/SchoolsSRA_chap3.html

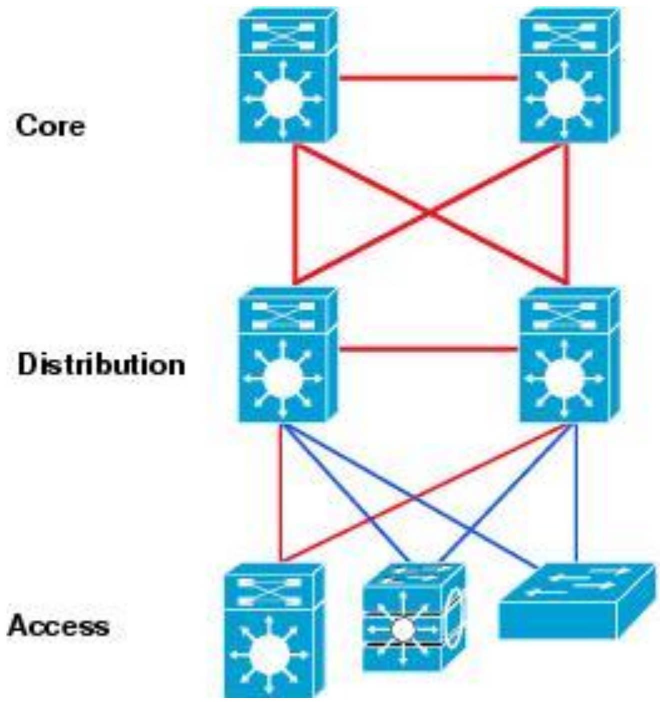

Figure 3. - 3-tier hierarchical model for high-performance, high-availability and scalable network design - Cisco

A VoIP interconnection agreement must include language that defines expected network performance in terms of availability and function. Service level requirements for each must be defined for all service-affecting parameters.

The Security Policy requirements, policies and practices subsumed in the operation of critical infrastructures, such as the PSTN, have been closely-guarded and proprietary to each PSTN participant. As a result, it would be impossible to conduct a trial (or report on resulting KPIs) of the security resiliency of a VoIP interconnection interface past its ability to withstand well-known attack scenarios because the nature of the attack, and how security has been breached, as well as a carrier's isolation and resolution procedures are highly-protected. However, under Executive Order, a cybersecurity framework has become the subject of recent efforts within the Department of Commerce National Institute of Standards and Technology (NIST). The goal is to develop a common framework for cybersecurity policies and practices.

Though these efforts are in early stages, it would benefit all service providers to remain apprised of (and implement) the recommendations and requirements resulting from these efforts. VoIP interconnection agreements should acknowledge this initiative and express the intention of both parties to comply with any reasonable efforts requiring cooperation between the parties to detect, isolate and resolve cybersecurity threats to the networks of either party.

NETWORK SUPPORT PRACTICES AND INFRASTRUCTURE INTER-OPERABILITY FOR EMERGENCY SERVICES

All networks require maintenance procedures and fault isolation procedures that may involve the facilities and equipment of interconnected carriers. Interconnection agreements for VoIP interconnection, therefore, should include provisions for joint TMN/FCAPS[23] efforts where warranted.

Infrastructure inter-operability for E911 services are well-defined for wireline carriers serving fixed-location subscribers. Because this document speaks to the migration of existing, fixed-location PSTN traffic and not all VoIP services in general (which may include nomadic VoIP services), the existing E911 Network practices for the wireline network should serve as the framework for infrastructure inter-operability using VoIP interconnection. To the extent nomadic VoIP service does traverse the VoIP interconnection point described in this document, the appropriate FCC rules for nomadic VoIP services would apply.[24]

NENA has developed a framework which encompasses proposed infrastructure and inter-operability standards for a future for NG9-1-1[25] service that would support multi-media (voice, text, video, graphics, etc.) information flows to Public Service Answering Points (PSAPs). Interconnection agreements for VoIP interconnection, therefore, should include provisions that assure each party can comply with governmental requirements for emergency services, and can exchange the required information flows across the VoIP interconnection interface for this purpose. An expressed intention by both parties to comply with the NENA standards for NG9-1-1 as required by state and/or federal regulatory agencies should provide the required certainty.

OSS PROCEDURES

Operational Support Systems represent the systems, processes and procedures necessary to forecast, order, provision, activate, manage, change, move, cancel or report (via Key Performance Indicators (KPIs)) facilities, equipment or functionality comprising carrier networks. Each carrier maintains its OSS to optimally serve the needs of its subscribers, product portfolios,

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[23] Telecommunications Management Network (TMN) is a term used to describe a separate network that has interfaces to the transport network for use in FCAPS efforts. FCAPS is an acronym meaning fault, configuration, administration (or accounting), performance and security. These are the management categories into which the OSI model defines network management tasks. See ITU-T Recommendations M.3010, M.3400, and X.700.

[24] See FCC First Report And Order And Notice Of Proposed Rulemaking, Adopted: May 19, 2005, Released: June 3, 2005 available at http://transition.fcc.gov/cgb/voip911order.pdf

[25] See NG9-1-1 Project description and associated documents including proposed standards available at http://www.nena.org/?NG911_Project

trading partners, management, government and other stakeholders. Because carrier networks must be interconnected, carriers must also interconnect at the OSS level. This allows the interconnected parties to ensure proper assignment of the inter-working components of each network necessary to support the operational profile required by the exchanged traffic at each point of interconnection.

Industry standards have been developed through the efforts of organizations such as the Ordering and Billing Forum (OBF) of the Alliance for Telecommunications Industry Solutions (ATIS). The interconnection agreement should specify the standards to be used for OSS inter-working as well as escalation procedures necessary to resolve process or procedural conflicts that may occur from time to time.

Interconnection Establishment/Activation Procedures

VoIP interconnection will supplant TDM interconnection over a period of time. In the case of a VoIP interconnection agreement between an RBOC and a competitive provider, this transitional period should be defined and scheduled. Such a schedule may consider time, traffic volume, traffic type, geographic area, points of interconnection or other useful method to organize the transition in totality. Using the information gathered in the steps outlined above, installation, acceptance testing and certification of the required media and signaling interfaces for each point of interconnection should be defined according to pre-established standards.[26]

The interconnection agreement should provide that, once acceptance testing has certified the points of interconnection, activation procedures will occur according to the schedule, and subject to the KPI measurement, management, oversight and reporting of service level covenants agreed upon and described more fully below.

Fallback Procedures

The interconnection agreement should specify that to the extent, during the activation process, any service level parameter is not or cannot be met for a legitimate reason, a remedy period within which the non-compliant party must resolve the issue is initiated and fallback procedures (to the pre-existing TDM interconnection framework) are immediately invoked. This will avoid service deterioration or outright network failure.

[26] Interconnection acceptance testing and certification should be conducted under mutually acceptable and documented inter-working standards such as ATIS-1000009.2006(R2011) and other associated standards.

KPI Measurement/Management/Oversight/Reporting of Service Level Covenants

Key Performance Indicators (KPIs) will form the foundation of service level covenants between parties to VoIP interconnection agreements. The methods used for management, oversight and reporting of KPIs, as well as the acceptable range for KPI values, should be specified for each service level covenant defined in the VoIP interconnection agreement. Further, if the measured parameter is found to be consistently outside of the acceptable operational range, the VoIP interconnection agreement should specify that the detailed performance of the components of the KPI should be reported to the interconnected parties. This further reporting requirement would aid both parties in identifying other areas that may be affected by the same service anomaly, and also in pursuing a possible joint resolution.

Acceptable voice (audio) quality, for example, is specified as a minimum MOS score of 4.0, as measured over a statistically valid sampling of traffic exchange. If the measured pMOS falls below this level for a statistically valid and extended period, the underlying cause for the quality erosion (ex. – excessive network delay, jitter, packet loss, etc.) may be the cumulative effect of specific, individual components of each party's network. The end-to-end resolution may require changes within both party's network regarding the routing path of traffic, the point of interconnection used, CODECs supported, etc. Sharing the specific component measurements of KPIs in such instances will help to preserve and maintain service quality.

Testing the Technical Elements of VoIP Interconnection is not Required

Because of the nature of the elements described above (each necessary to affect a ubiquitous and functional VoIP interconnection framework for the PSTN), a trial conducted for the purpose of testing all in combination or even each of them individually is simply not possible or practical. For example, as mentioned earlier, adopting a *minimum* list of available interconnection points could facilitate negotiation (although merely finding that the statutory mandate that any technically feasible point of interconnection is available would provide all the guidance that good faith negotiations requires). Conversely, a trial of the different physical network interface technologies available for VoIP interconnection is unnecessary, since those technologies are well understood and in common use today for interexchange services.[27]

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[27] In fact, the Alliance for Telecommunications Industry Solutions has created the Next Generation Interconnection Interoperability Forum (NGIIF), which has driven the creation of a number of standards and practices to facilitate VoIP interconnection. According to ATIS: "The NGIIF addresses next-generation network interconnection and interoperability issues associated with emerging technologies. Specifically, it develops operational procedures which involve the network aspects of architecture, disaster preparedness, installation, maintenance, management, reliability, routing, security, and testing between network operators. In addition, the NGIIF addresses issues which impact the interconnection of existing and next generation networks and facilitate the transition to emerging technologies."

Further, the functional acceptability, by certificated PSTN carriers, of Voice over Internet Protocol as a technology, is not in dispute and has been operationally evident for more than a decade. Therefore, a “technical” trial to confirm what the industry already knows (that it works) is not useful. Finally, even the technical requirements for inter-carrier network-to-network interfaces supporting VoIP have been documented as an American National Standard for Telecommunications[28] since May of 2006, albeit the evidence indicates that the RBOCs have been reticent, for the past seven years, to use them.

What may ultimately be useful, however, is testing the administrative and operational procedures and practices, already resident or yet to be created within the Operational Support Systems of the interconnected providers that will enable the orderly transition of the PSTN to VoIP technology. TMN/FCAPS process inter-operability as well as inter-operability of infrastructure supporting emergency services may also benefit from testing in the future, as new databases come online. The future test could comprise plans to validate FCAPS processes and competitive access to the E911/NG911 Network serving appropriate PSAPs, in scenarios where PSAPs for multiple municipal jurisdictions are accommodated through a common VoIP interconnection point.[29] However, it is virtually impossible to develop test plans or test scenarios of even these OSS processes and procedures without a clear understanding of the underlying responsibilities of each party in a production environment that occurs after an interconnection agreement has been negotiated.

Conclusion

The first step in achieving the transition of the PSTN to IP is VoIP interconnection and transport. In order to initiate this transition, the regulatory framework must first be confirmed as falling under section 251/252 of the Telecommunications Act. Thereafter, the interconnection agreements governing the traffic exchange between competitors and RBOCs will require expansion to include the technical parameters necessary to support VoIP interconnection, similar to those already specified for TDM interconnection in such agreements. These parameters are well understood, because of the operational experience of more than a decade garnered by the majority of carriers that already exchange interexchange voice traffic in IP format. Therefore, with agreements in place, nothing of a technical nature should preclude the swift and orderly completion of this first step in the transition of the PSTN to IP.

[28] See ATIS-1000009.2006 (R2011) “IP Network-to-Network Interface (NNI) Standard for VoIP” available at https://www.atis.org/docstore/product.aspx?id=25486

[29] Of course, FCAPS and E911 testing are always fully incorporated into the general acceptance testing performed before any POI is put into production. A VoIP POI would enjoy no exception to this procedure.